SPARK

Software Modulator for
Digital Radio Mondiale (DRM)

User Manual

Version:	1.0
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Last changed:	16th August 2009
First published:	07th July 2009

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Abbreviations

AAC	Advanced Audio Coding
AGC	Automatic Gain Control/Correction
AM	Amplitude Modulation
AMSS	Amplitude Modulation Signalling System
BIOS	Basic Input and Output System
DAQ	Data Acquisition
DCP	Distribution and Communication Protocol
DRM	Digital Radio Mondiale
EEP	Equal Error Protection
ETSI	European Telecommunications Standards Institute
FAC	Fast Access Channel
FEC	Forward Error Correction
FFT	Fast Fourier Transform
FIR	Finite Impulse Response
GMT	Greenwich Mean Time
HmMix	Mixed Hierarchical Mapping
HmSym	Symmetrical Hierarchical Mapping
HPP	Higher Protected Part
IF	Intermediate Frequency
ISI	Inter-Symbol-Interference
JRE	Java Runtime Environment
JVM	Java Virtual Machine
LGPL	Lesser General Public License
LPP	Lower Protected Part
MDI	Multiplex Distribution Interface
MLC	Multi-Level Coding
MOT	Multimedia Object Transfer
MPEG	Motion Experts Picture Group
MSC	Main Service Channel
NTP	Network Time Protocol
OFDM	Orthogonal Frequency Division Multiplexing
PCM	Pulse Code Modulation
PFI	Programmable Function Interface
PFT	Protection Fragmentation and Transportation

PRBS	Pseudo Random Binary Sequence
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
RF	Radio Frequency
SBR	Spectral Bandwidth Replication
SDC	Service Description Channel
SDR	Software Defined Radio
SM	Standard Mapping
TF	Transmission Frame
TSF	Transmission Super Frame
UDP	User Datagram Protocol
UEP	Unequal Error Protection
UTF	Unicode Transformation Format
VHF	Very High Frequency
VSPP	Very Strongly Protected Part
XOR	Exclusive Or

1 Introduction to DRM

In 1998 the Digital Radio Mondiale (DRM) consortium started to develop a European standard for digital sound broadcasting in the long, medium and short wave bands [1]. The first version of the standard was published in 2003 by the European Telecommunications Standards Institute (ETSI). The DRM system was designed for worldwide audio and data broadcasting in frequency domains below 30 MHz and for channel bandwidths from 4.5 kHz to 20 kHz. In 2004 the consortium members proposed to extend the standard by introducing a new transmission mode which allows for broadcasting in the very high frequency (VHF) regions. The name of the new standard was chosen to be Digital Radio Mondiale Plus (DRM^+).

1.1 The DRM Transmission Chain

A schematic overview of the DRM transmission chain is shown in Figure 1.1.

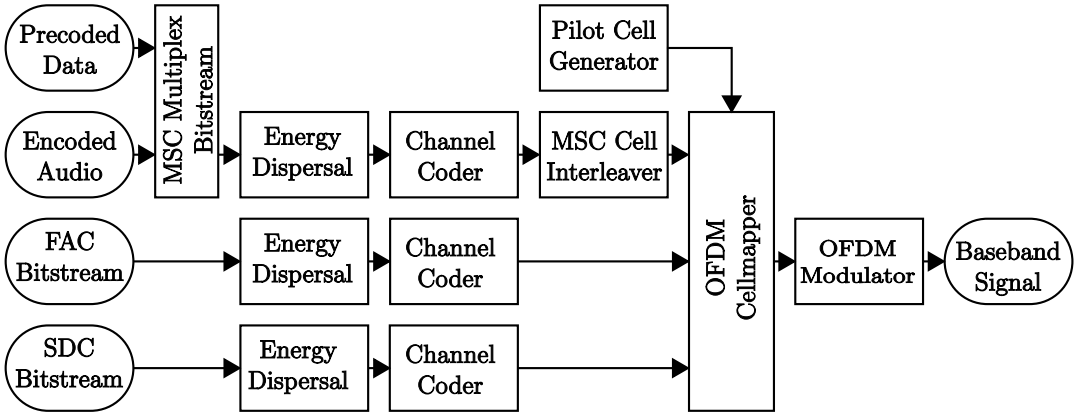

Figure 1.1: Signal flow graph of a DRM transmission chain [1].

The DRM transmission chain consists of three data channels: The fast access channel (FAC), the service description channel (SDC), and the main service channel (MSC). In the following, the most important parts of the transmission chain are explained. Since the knowledge of the channel coding and modulation

schemes is required in order to depict the properties of the DRM data channels, the explanation begins with the description of the energy dispersal stage.

1.2 Energy Dispersal

Before channel coding, each bitstream is processed by a scrambler stage. The XOR pattern for the scrambler is derived from a pseudo random binary sequence (PRBS) generator with generator polynomial $x^9 + x^5 + 1$.

1.3 Channel Coding

The scrambled bits are forwarded to a convolutional encoder with a mother code rate of 1/6 and 6 bit constraint length, where the generated code is terminated by an all-zero input sequence. Different puncturing patterns can be applied in order to select the desired output code rate. Additionally, tail-bit-puncturing is used. For audio or data content, DRM supports four different code rates which can be chosen from a *protection level* table in order to meet the coding gain requirements. After channel coding, the bits are interleaved over one frame by a convolutional interleaver and forwarded to the modulation stage.

1.4 Orthogonal Frequency Division Multiplexing (OFDM)

DRM uses OFDM, a multicarrier modulation scheme that is widely used in modern digital broadcasting applications. The idea of multicarrier transmission is to first split the multiplex bitstream into different substreams and then to encode these streams in parallel, e.g. by using multilevel coding (MLC). Subsequently, the bits of each individual substream are mapped onto one of the N_C subcarriers by shaping the carrier's amplitude and phase according to a predefined mapping function. Different mapping functions and different *constellation* types are available.

1.5 OFDM Framing

The OFDM framing in DRM is strictly hierarchical. The top level framing unit is the transmission super frame (TSF) which contains a sequence of transmission frames (TF). Each transmission frame wraps N_{SYM} OFDM symbols.

1.6 The DRM Data Channels

The internal data structure of DRM is depicted in Figure 1.1. There are three different channels serving different purposes, each of them is briefly explained in the following subsections.

1.6.1 Fast Access Channel (FAC)

The FAC cells are included in every transmission frame and carry information about the OFDM configuration and other system settings. The fixed code rate and the position of the QPSK symbols close to the reference cells make the FAC very robust. Reliable access to the data of the FAC is necessary to provide the receiver with the information needed to decode the SDC and MSC.

1.6.2 Service Description Channel (SDC)

The SDC contains information on the streams and services, included in the MSC multiplex. In DRM mode the SDC can be configured to use either 4 QAM or 16 QAM modulation and a fixed code rate of $r = 0.5$. In DRM$^+$ mode the SDC can be configured to use the code rates $r = 0.5$ or $r = 0.25$ but the modulation is fixed to 4 QAM. In both modes the encoded cells are mapped onto the first couple OFDM symbols within a TSF.

1.6.3 Main Service Channel (MSC)

The MSC carries the precoded and multiplexed digital media content. In DRM$^+$ mode the constellation mapping can be chosen to be either 4 or 16 QAM together with different code rates. In DRM mode, the MSC modulation is restricted to 16 QAM or 64 QAM. The MSC allows for the use of unequal error protection (UEP), where the multiplex frame is split into higher and lower protected parts by using separate code rates for each respective part[1]. Additionally, the MSC uses a convolutional QAM cell interleaver, which makes the main service channel less vulnerable to multi-path fading.

[1] In the DRM standard the available code rates for the higher and lower protected parts are subdivided into four different protection levels.

2 Introduction to Spark

2.1 Functionality

Spark is a realtime software modulator for DRM, DRM$^+$and AM. It combines DRM content server and modulator capabilities in one software defined radio (SDR) application. Additionally, Spark supports the multiplex distribution interface (MDI) which allows to broadcast the multiplex content over ethernet.

Along with the software, a graphical user interface is provided for setting up the DRM signal parameters (e.g. audio bit rate, transmission mode, spectrum occupancy), the stream sources and destinations (e.g. sound card or file) and the service information (e.g. Station Label).

2.1.1 Audio Streams

Spark is able to acquire audio signals from the line input of a PC soundcard, from a wave file of the format (RIFF, 16 bit, 48 kHz, Stereo) or from MP3 files. For testing purposes, the LGPL open-source AAC library FAAC can be used which does not support stereo or SBR transmissions. For full AAC+ audio coding a "DRM Audio Encoder Library Module" for MPEG AAC+SBR coding available from Dolby, Deutschherrnstr. 15-19, D-90429 Nuernberg, Germany, (http://www.dolby.com) can be licensed. The software library can be licensed to the client through the developer which has an according contract with Dolby.

DRM text messages can be entered directly using the PC keyboard in the input dialogue of the audio stream. A number of text messages can be sent cyclically (text carousel) with an adjustable repetition for each message. Text message reconfiguration during transmission is supported.

2.1.2 Data Streams

The software supports the multimedia object transfer (MOT) data protocol. The supported applications are MOT Broadcast Website and MOT Slideshow. Additionally, Spark supports the transmission of pseudo random binary sequences (PRBS) as specified in [2]. The files for MOT operation can be read from the drives of the operating system.

2.1.3 Time Synchronisation

The software automatically acquires the time by one of the network time protocol (NTP) servers specified in the time input dialog during modulation. If there is no NTP server available, the BIOS clock of the PC on which the software is running can be used.

2.1.4 Output

The DRM multiplex can either be modulated or send over an IP network using the MDI. The MDI protocol specification version 1.1.1 is fully implemented. However, the distribution and communications protocol (DCP) does not support Reed-Solomon forward error correction (FEC).

2.1.5 Realtime Operation

Realtime operation of the software is possible. Due to internal buffering, the overall delay between the analogue audio input and the DRM signal output will be in the order of 3 to 5 seconds.

2.1.6 Reliability

The software is stable and has been tested in a permanent setup without any indications of instability or other problems. The stability of the Java virtual machine running on the host PC cannot be guaranteed.

2.1.7 Restrictions

The following restrictions on the functionality and the operation of the software apply:

- Up to four audio streams can be used can be used for generation of the DRM signal. The DRM Multiplex is static, reconfiguration during operation is not supported (reconfiguration index = 0).
- Only data entities of the types 0, 1, 5, 8, 9, 10, 12 are generated.
- For audio coding, only AAC is possible.
- The decodability of the DRM signal output of Spark was successfully verified with the Morphy Richards DRM receiver, the Fraunhofer Softwareradio v4.0.4, the DIORAMA software receiver and the DREAM software receiver version 1.1.4. The MDI interface and MDI transmitter application was tested with the Fraunhofer (FhG) Softwareradio. However, the full compliance according to [1] can not be guarantueed.
- The jitter and frequency accuracy of the analogue output signal is limited by the output hardware. This may lead to a situation where the RF frequency accuracy required by frequency managing authorities may not be met.

2.2 Requirements

2.2.1 Hardware requirements

The hardware requirements for Spark strongly depend on the configuration of the software. The minimum hardware requirements for Spark are fulfilled by a PC with an 800 MHz CPU and with 128 MB of RAM dedicated to the applicaten. Given these constraints Spark will run stable in DRM mode at output sample rates of not more than 48 kHz and not more than 2 AAC streams. However, the user might observe signal dropouts as well as insufficient input delays.

In order to run Spark stable at sample rates above 48 kHz, a PC with a 2.0 GHz dual core CPU togther with a minimum of 512 MB RAM reserved for the application is recommended.

2.2.2 Software requirements

The Spark user interface as well as the baseband algorithms are written in Java and require the Java Virtual Machine (JVM) which is included in the Java Runtime Environment (JRE). Spark has been developed to work with the JRE version 6.0 or above. Note that it is not recommended to use JRE versions prior to JRE 1.6 update 13 together with the software.

Since the main components of Spark are written in Java, the application is in general platform independent. However, due to the native library support, which is used for the FFT, the AAC encoder and the native input and output devices, the range of operating systems to run Spark on is limited to Microsoft Windows and Linux platforms. Although it is possible to provide native support for other operating systems, the support for third party libraries (e.g. Dolby AAC+) can not be guarantueed.

3 Spark User Manual

3.1 The Application Window

After Spark has been started and preconfigured the **main window** is shown as depicted in Figure 3.1.

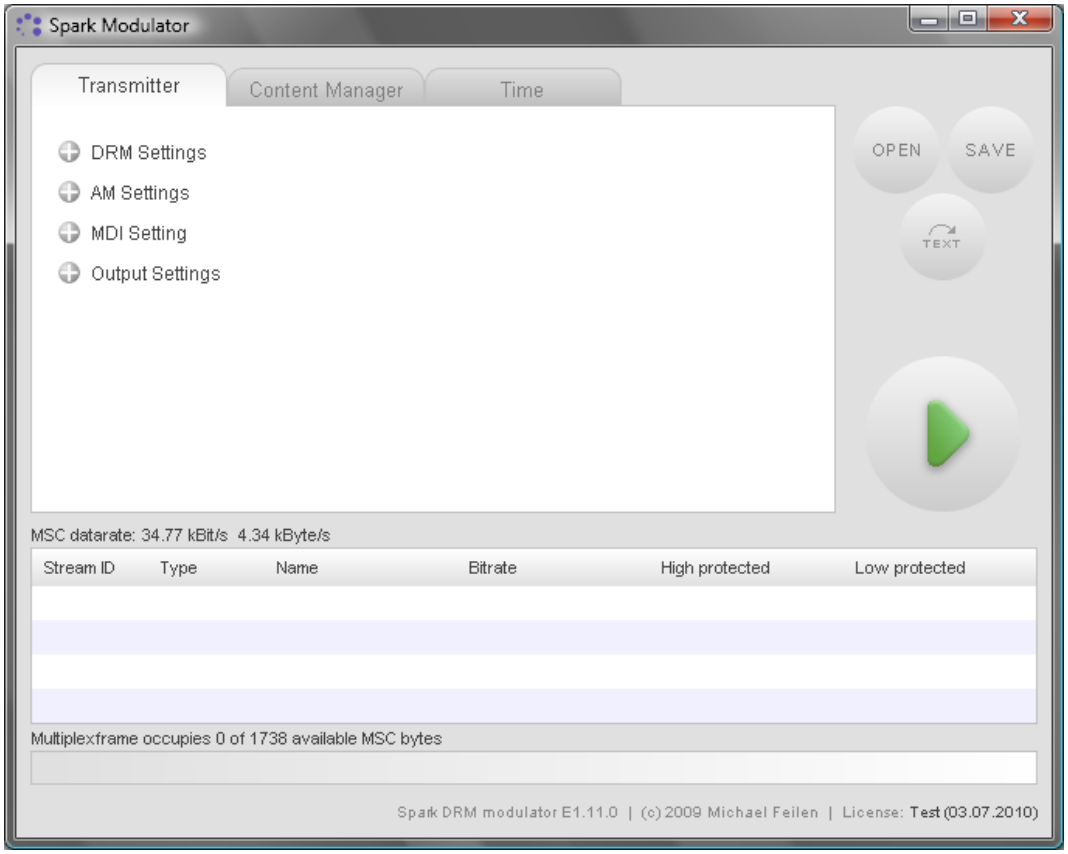

Figure 3.1: The Spark application window.

The **configuration window** in the center of the screen comprises of three tabs: the transmitter tab, the content manager tab and the time tab. The functionality of each tab will be explained in the following:

- **Transmitter tab:** In the transmitter configuration panel, the transmission mode and the output devices can be configured. Details on the transmitter settings can be found in Section 3.2.

- **Content Manager tab:** The content manager contains the stream and service information together with the input device configuration. The content manager settings are explained in Section 3.3.

- **Time tab:** The time configuration window contains the NTP time reference configuration and other time reference related settings. The time reference related parameters are discussed in Section 3.4.

The settings of the panels in the different tabs can be saved and loaded by the oval open and save buttons in the **control panel** on the top-right hand side of the main window. The oval button with the green arrow in the control panel can be used to start or stop the transmitter, i.e. the DRM$^+$, AM modulation or the MDI broadcast. Additionally, the control panel contains a text-button for text message reconfiguration. This button becomes active after the transmission has been started and is inactive as long as the modulator is idle. Detailed information on the control panel functionalities can be found in Section 3.5.

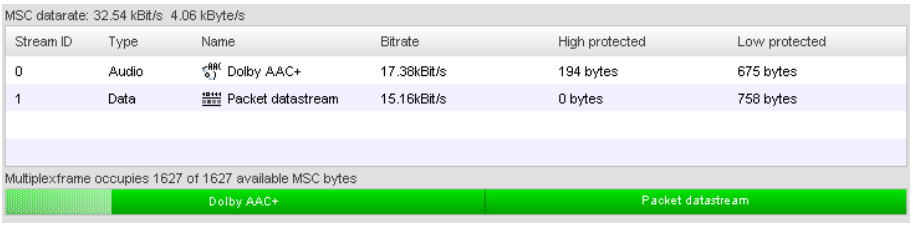

Figure 3.2: The multiplex information panel with a content manager example configuration.

On the bottom, the main window contains the **multiplexer panel** which gives an overview of the available multiplexer bandwidth for the current content manager configuration and which furthermore indicates the bandwidth consumption of the streams that have been added to the content manager. An example configuration of the multiplex bandwidth information panel is shown in Figure 3.2.

In the following sections the available configuration options and the different configuration parts are explained.

3.2 The Transmitter Settings

The transmitter settings tab contains the configuration parameters for the modulation and the output devices. It comprises of the four different submenus:

- **DRM Settings:** The DRM settings submenu contains the configuration options for the DRM OFDM configuration and the DRM channel coding configuration for the channels MSC and SDC.
- **AM Settings:** The AM settings submenu contains the configuration options for the AM transmission.
- **MDI Settings:** The MDI settings submenu contains the options for the DRM MDI transmission over IP.
- **Output Settings:** In the output settings submenu the transmission mode can be configured as well as the PCM output devices for OFDM transmission.

3.2.1 DRM Settings

Figure 3.3 shows the available parameters in the DRM settings input box. In the following each parameter is briefly explained.

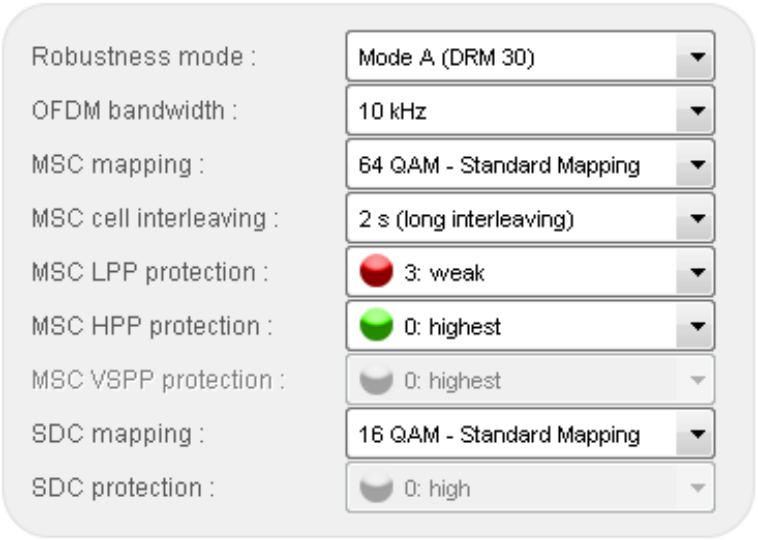

Figure 3.3: DRM settings input box.

Robustness Mode and OFDM bandwidth

The **robustness mode** defines the OFDM subcarrier spacing and the OFDM pilot configuration [1]. The spectrum occupancy of the OFDM signal is defined as **OFDM bandwidth**. The higher the OFDM subcarrier spacing and the higher the number of pilot cells, the higher the robustness of the transmission but the lower the available MSC datarate.

In DRM there exist five different robustness modes, whereas each mode is compatible to a different set of OFDM bandwidths. Therefore, a reconfiguration of the robustness mode may force a reconfiguration of the OFDM bandwidth parameter. Changing the robustness mode may also induce a **DRM mode** change as the robustness modes A,B,C and D are specified for DRM up to 30 MHz and robustness mode E defines a DRM$^+$ transmission in the VHF region. A detailed overview of the different robustness modes and OFDM bandwidth configurations is given in Table 3.1.

OFDM bandwidth [Hz]	Robustness mode	DRM mode
4,500	A,B	DRM
5,000	A,B	DRM
9,000	A,B	DRM
10,000	A,B,C,D	DRM
18,000	A,B	DRM
20,000	A,B,C,D	DRM
100,000	E	DRM$^+$

Table 3.1: OFDM bandwidth and the supported DRM robustness modes.

SDC and MSC mapping

The parameters for **MSC mapping** and **SDC mapping** define the mapping of the multi-level channel coder (MLC) output bits onto a QAM cell for each respective channel. The acronym QAM stands for quadrature amplitude modulation, where the phase and the amplitude of the transmitted carrier are shaped by a certain mapping function as stated in Section 1.4.

In DRM it is distinguished between *standard mapping* (SM) and hierarchical mapping, where the hierarchical mapping is subpartitioned into *mixed hierarchical mapping* (HmMix) and *symmetrical hierarchical mapping* (HmSym). When HmSym is used hierarchical modulation is performed on both, the inphasal (I)

and the quadrature (Q) component, respectively, whereas in case of HmMix only the inphasal component is modulated hierarchically. MSC hierarchical mapping is available only in 64 QAM mode. The available MSC and SDC mapping schemes are listed in Table 3.2 and Table 3.3, respectively.

NOTE: When hierarchical mapping is used the data of the *first stream* in the multiplexer (stream ID equal to zero) is transmitted in the so called hierarchical data part. Therefore it is important that the number of bits for the multiplex stream with ID 0 is less or equal to the number of bits available for the hierarchical part. Furthermore, the hierarchical data part must not have a higher protected part, i.e. the use of UEP in the stream with ID 0 is permitted.

Mapping scheme	Resolution	DRM mode
Standard Mapping	4 QAM	DRM^+
Standard Mapping	16 QAM	DRM, DRM^+
Standard Mapping	64 QAM	DRM
Symmetrical Hierarchical Mapping (I and Q)	64 QAM	DRM
Mixed Hierarchical Mapping (I only)	64 QAM	DRM

Table 3.2: MSC mapping schemes.

Mapping scheme	Resolution	DRM mode
Standard Mapping	4 QAM	DRM, DRM^+
Standard Mapping	16 QAM	DRM

Table 3.3: SDC mapping schemes.

MSC cellinterleaving

The MSC QAM cells are interleaved over a certain number of transmission frames in order to increase the signal robustness in case of burst errors, e.g in a channel environment with flat multipath fading. The higher the interleaver depth, the longer the delay between signal acquisition and the audio playback at the receiver. Table 3.4 summarizes the available MSC cell interleaver parameters subject to the different DRM modes.

Interleaver depth	Duration [ms]	DRM mode
1 (short)	400	DRM
5 (long)	2000	DRM
6	600	DRM$^+$

Table 3.4: Interleaver configuration.

MSC and SDC protection

The protection level reflects the code rate for the respective part of the MSC or the SDC bitstream. The lower the code rate, the more redundancy information is added in the process of channel coding. Although a smaller code rate increases the robustness of the transmitted information it decreases the number of usable information bits. Concerning the MSC, DRM distinguishes between the lower protected part (LPP), the higher protected part (HPP) and the very strongly protected part (VSPP). If the protection levels for LPP and HPP are equal, the MSC is said to use equal error protection (EEP). If the protection levels are different the MSC is said to use unequal error protection (UEP). The VSPP protection level defines the coderate for the hierarchical data part. Since the VSPP is only present when hierarchical modulation is used, the field is disabled when using standard mapping for the MSC.

NOTE: Different protection levels for the SDC are only available in DRM$^+$ mode, i.e. when using robustness mode E.

Spark displays the protection level as a number from zero to four as defined in the DRM specification [1] followed by one of the following robustness indicators: *highest*, *high*, *medium* and *weak*.

3.2.2 AM Settings

The AM settings panel contains the configuration for the amplitude modulation functionality of Spark. The available input options in the AM settings panel are shown in Figure 3.4.

The user can specify the audio bandwidth as well as the modulation mode and modulation degree.

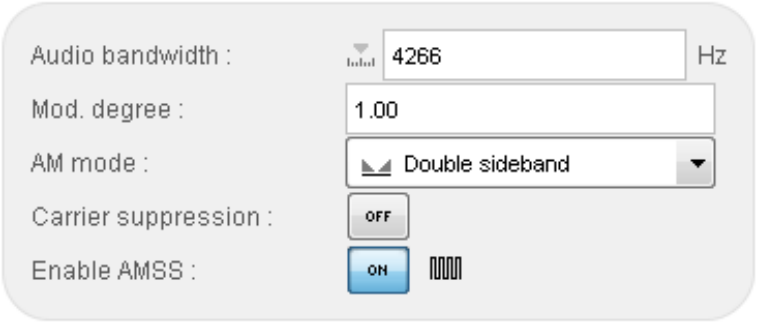

Figure 3.4: AM settings input box.

Audio bandwidth

The audio bandwidth can be adjusted according to the AM channel bandwidth, which is typically 4.5 or 5 kHz.

Modulation degree

The modulation degree indicates the ratio of the peak modulation signal amplitude, i.e. the maximum audio signal amplitude, to the peak amplitude of the carrier. If the peak amplitude of the modulation signal is half the carrier amplitude, the modulation degree is 0.5. If the peak amplitudes are equal, the modulation degree is 1. In Spark, the modulation degree can be adjusted between 0.1 and 1.0.

AM mode

The AM mode describes the position of the output spectrum relative to the carrier frequency. In Spark, the user can choose between *lower sideband*, *upper sideband* and *double sideband* modulation. Lower sideband means the output spectrum is located at the left hand side of the carrier. Upper sideband indicates that the position of the output spectrum is at the right hand side of the modulation frequency. In case of double sideband modulation the spectrum is symmetrical around both sides of the carrier frequency.

Carrier suppression

To achieve a better output power efficiency carrier suppression can be enabled. When carrier suppression is enabled the power of the carrier is reduced by approximately 12 dB.

Enable AMSS

When the amplitude modulation signalling system (AMSS) is enabled, the service information such as the service label, the service identifier and other information is send by phase modulation of the AM carrier. For more information on AMSS please refer to [3].

3.2.3 MDI settings

The MDI settings panel contains the configuration options for the distribution and communication protocol (DCP) [4] and other protocol-specific settings.

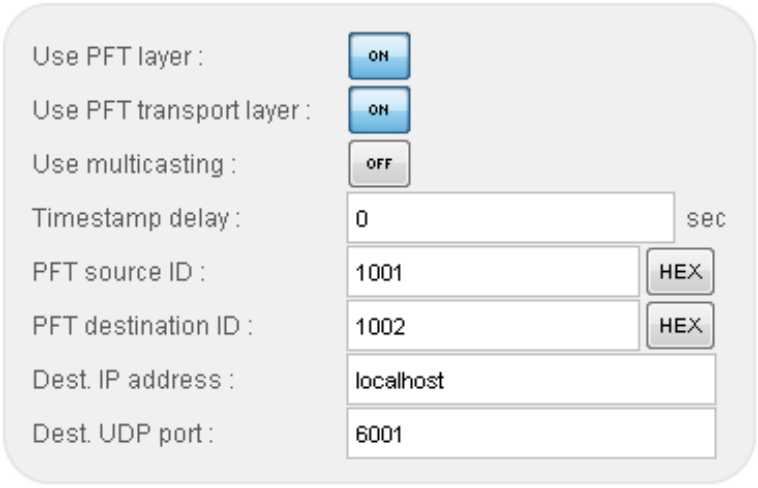

Figure 3.5: MDI settings input box.

Use PFT layer

The protection, fragmentation and transportation (PFT) layer is used for data framing, data encoding as well as source and destination addressing. If the use PFT layer button is disabled, the MDI communication carried out on the AF layer only.

Use PFT transport layer

The PFT layer includes a transport layer where the packet source (transmitter) can be identified by a 16 bit source ID and a packet destination (receiver) can be identified by a 16 bit destination ID. In a network with multiple MDI transmitters

and receivers this enables to specifically address one particular MDI packet destination and to identify one particular MDI packet source. Disabling the transport layer button disables this addressing feature.

Use multicasting

If the *Use multicasting* button is enabled the MDI packets can be send to a group of MDI receivers. Note that in case of multicast transmissions the *Dest. IP address* input field must contain a multicast IP address within the address range 224.0.0.0 to 239.255.255.255 (IPv4).

Timestamp delay

The timestamp delay defines the number of seconds added to each timestamp that is generated and transmitted in the MDI output stream. The timestamp is generated by the time reference (e.g. NTP) which is specified in the time configuration tab of the configuration tab panel. For a detailed overview of the MDI timestamp functionality please refer to [5].

PFT source ID

The PFT layer source ID of the MDI packet source (transmitter) must be in the range of 0 to 65535 (see *Use PFT transport layer*).

PFT destination ID

The PFT layer destination ID of the MDI packet sink (receiver) must be in the range of 0 to 65535 (see *Use PFT transport layer*).

Destination IP address

The IP address of the MDI packet sink (receiver) must be entered in the *Dest. IP address* input field.

Destination UDP port

The UDP port number for the DCP-MDI packet communication (default: 6001) must be entered in the *Dest. UDP port* input field.

3.2.4 Output Settings

The output settings dialog contains the settings for the PCM output devices, the transmission mode, the settings for the OFDM signal generation and the modulation settings as shown in Figure 3.6.

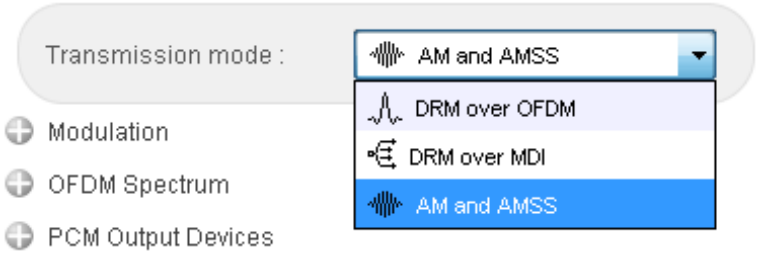

Figure 3.6: Output settings input box.

Transmission mode

The transmission mode defines the output operation of Spark. The user can choose between the following output modes:

Output mode	Description
DRM over OFDM	Modulated DRM OFDM signal output over a PCM output device, e.g. a soundcard.
DRM over MDI	DRM data packets transmitted over ethernet by using the MDI interface.
AM and AMSS	Amplitude modulated audio signal output over a PCM output device, e.g. soundcard.

Table 3.5: Spark output transmission modes.

3.2.5 Output Settings / Modulation

The modulation panel contains the output settings for the output modulator as depicted in Figure 3.7. The modulation settings only affect the OFDM and AM transmission modes.

Figure 3.7: The modulation panel in the output settings tree.

Intermediate frequency (IF)

The intermediate frequency defines the frequency in Hz to which the complex baseband signal is shifted after modulation.

Output frequency

The output frequency defines the frequency in Hz to which the real PCM output IF signal is shifted by an external hardware device. This box is disabled if no external device (e.g a DiRaGen modulator) was found.

Invert baseband spectrum

If the *Invert baseband spec.* button is enabled the baseband spectrum will be inverted before modulation. Hence, the new complex baseband signal is the complex conjugate of the original complex baseband signal.

3.2.6 Output Settings / OFDM Postprocessing

FIR filtering

If the *FIR filtering* button is enabled the FIR output filter will be used to shape the OFDM signal in order to reduce the sideband emissions. The FIR filter parameters are calculated using the Kaiser windowed sinc design method.
NOTE: Using the FIR filter causes inter-symbol-interference (ISI). However, the influence of ISI is negligible in most cases.

Sideband suppression

The sideband suppression defines the attenuation of the output signal from the beginning to the end of the transition region. The higher the sideband suppression the more filter coefficients are required for adequate image rejection and the

higher the computational complexity.
Recommendation for DRM and DRM$^+$: 50 dB

Transition bandwidth

The transition bandwidth parameter modifies the steepness of the filter transition region and is given in Hz. A small transition bandwidth requires a higher filter order but gives a steeper sideband rolloff.
Recommendation for DRM: 500 Hz
Recommendation for DRM$^+$: 2500 Hz

FIR coefficient file

In the *FIR coefficient file* input field an ASCII file can be specified, which must contain the FIR filter coefficients for the OFDM output signal filtering. If no file is specified, Spark designs an adequate OFDM signal filter.

The coefficients in the file must be separated by a newline character (\n). Comments in the file must start with two forwardslashes (//) and must not be placed before coefficients.

Example of an FIR coefficient file:

```
// This is a comment
-0.000000228647259
 0.000000925740866
-0.000001940961649
-0.000002204589717
 0.000001358034607
...
-0.000000228647259
```

Enable output AGC

The automatic gain correction (AGC) algorithm scales the output signal with respect to the *Gain* value, specified in the output device input panel, in order to reach the maximum PCM output device resolution. When the output AGC is disabled the OFDM signal power per carrier is equal to power given in the DRM specification, i.e. the gain value of the output device is ignored.

3.2.7 Output Settings / PCM Output Devices

In this subsection the different PCM output devices can be configured. Furthermore, the user can choose the current output device as well as the output buffersize.
For each output device the output gain, the number of channels, the sample resolution in bits per sample and the sample rate can be specified.

PCM output device

The user can choose one output device from a list of output devices as listed in Table 3.6.

PCM output device	Clock reference
Line Out (Soundcard)	Soundcard crystal
Wave File	CMOS Timer
NiDAQmx	External (PFI0) or DAQ card crystal
IQ via UDP	CMOS Timer
Native Device	User defined

Table 3.6: Available PCM output devices.

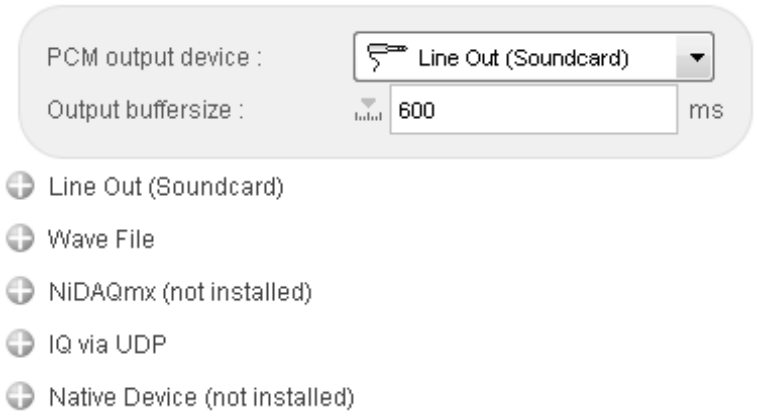

Figure 3.8: Analog output device input box.

3.3 Content Manager Configuration

The content manager multiplexes the data provided by the different MSC streams and manages the services which are assigned to the streams and their particular

substreams.

The content manager configuration is shown in the **Content Manager** tab. An example configuration of the content manager panel showing one AAC stream and one service with the label "Spark" is shown in Figure 3.9. The panel gives an overview of the service (blue), stream (green) and substream (yellow) configuration and allows the user to change the respective parameters.

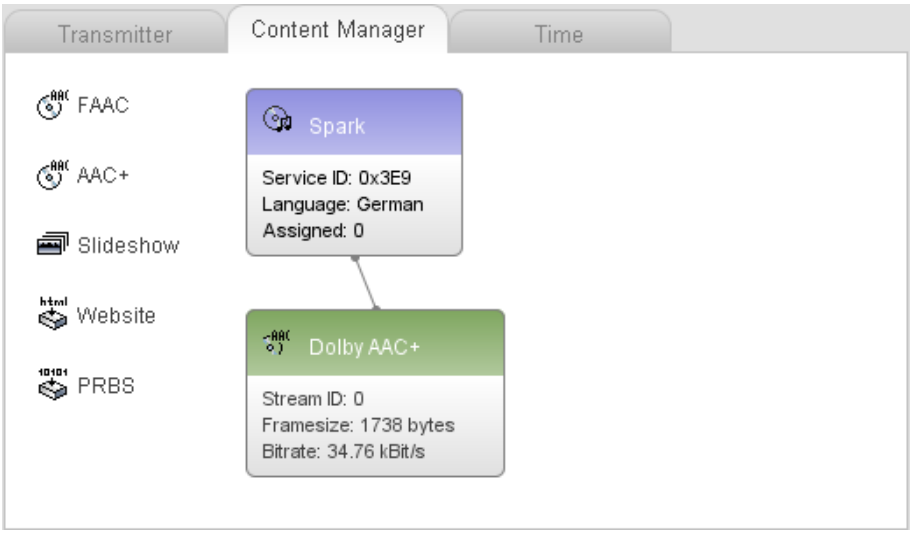

Figure 3.9: Content manager tab showing an example AAC configuration and the buttons to create new audio or data streams.

3.3.1 Introduction to streams and substreams

In DRM a logical stream is the low layer representation of data or audio information. The number of usable streams and substreams is limited to a maximum number of four streams. The streams are embedded in the DRM multiplex and are uniquely identified by a *stream ID* which is an integer number in the range of 0 to 3. DRM distinguishes between audio and data streams. As far as datastreams are concerned it is distinguished between asynchronous and synchronous datastreams [6].

To create a new stream the user must pick one stream from the list on the left hand side of the content manager panel as shown in Figure 3.9. Table 3.7 lists the streams available in Spark and explains their functionality and service type classification.

Stream type	Description	Service type
FAAC	Monaural AAC audio stream without spectral bandwidth replication (SBR) and without parametric stereo (PS).	Audio
AAC+	Licensed DRM$^+$ Dolby AAC+ audio stream supporting SBR and PS.	Audio
Slideshow	The slideshow substream allows the cyclic transmission of pictures in various formats (e.g. JPEG, GIF, BMP, etc.) and is carried by a packet data stream.	Data
Website	The files that are required to render a website on the receiver must be placed in a certain directory. The contents of this directory can then be transmitted using the MOT website substream which is carried by a packet data stream. The user must define an entry page which indicates the first page the receiver is advised to display (e.g. an index page of a website).	Data
PRBS	The pseudo random binary sequence can be transmitted in synchronous or asynchronous mode. The PRBS generator polynomial is defined in [2]. Since the PRBS constitutes a predictable bit sequence the PRBS stream can be used to evaluate the bit error rate of a DRM$^+$ receiver.	Data

Table 3.7: Available audio and data streams in Spark.

After one of the stream buttons on the left hand side of the content manager has been pressed, the user is prompted to define a bitrate for the new stream in the bitrate suggestion dialog as shown in Figure 3.10. Although it is required to choose a bitrate from the bitrate suggestion dialog, the selected bitrate can be freely adjusted later in the stream configuration panel.

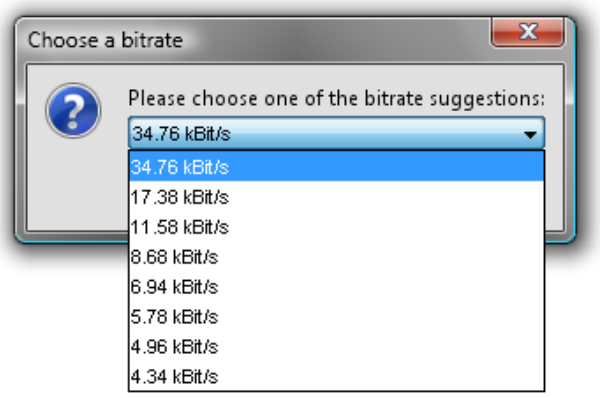

Figure 3.10: The bitrate suggestion dialog appears before a new stream is created.

After the bitrate has been chosen, the configuration panel of a stream appears. The stream settings can be changed anytime by clicking on the stream in the content manager panel and selecting *Edit* from the popup menu as shown in Figure 3.11.

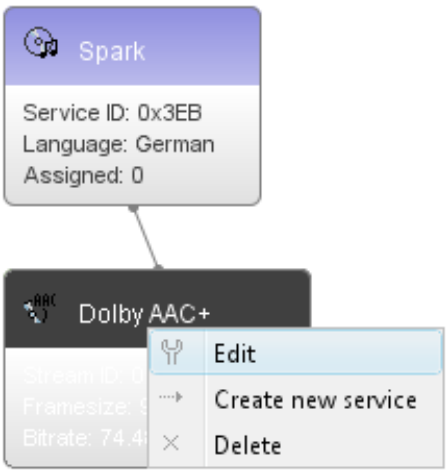

Figure 3.11: Opening the configuration dialog of a stream.

3.3.2 AAC Stream

Clicking on the *Edit* button in the popup menu lets the AAC stream configuration dialog appear as shown in Figure 3.12. AAC stream configuration dialog is subdivided into three menus:

- Audio Input Device

- Stream Datarate

- Text Messages

In the following the functionality and options of each submenu are explained.

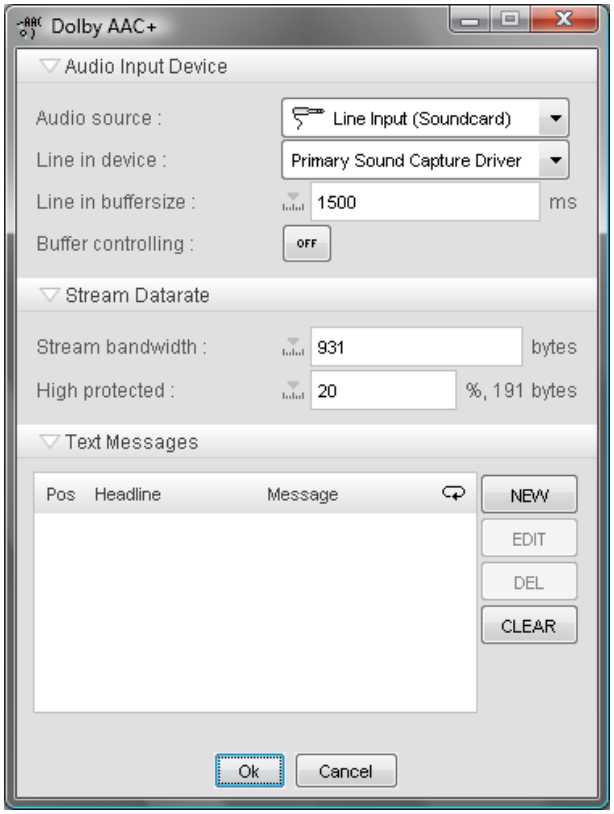

Figure 3.12: AAC stream configuration dialog.

Audio Input Device

Table 3.8 shows the available audio input devices for an AAC stream in Spark.

Device	Description
Line Input (Soundcard)	Audio which is sampled in realtime from the device specified in the *Line in device* box. The selected soundcard must support 16 bit, 2 channels and a samplerate of 48 kHz. If *Buffer controlling* is enabled, the audio input data is processed by a resampler which synchronizes the input device sample clock to the output device sample clock to prevent the audio input buffer from overflowing or underrunning.
Wave File	Reads sampled audio from a .wav file. The wave file sample format is required to be 16 bit and 48 kHz. If the *Repeat* button is enabled, the wave file is played in a loop.
MP3 Playlist	The user can add files and directories to an MP3 playlist. The files are played in the order in which they appear in the list. When pressing the *Add files* or the *Add folder* button the file chooser dialog appears. If *Scan subdirectories* is enabled, the subdirectories of the folder selected via the *Add folder* option are scanned for files with ".mp3" suffix and the respective files are added to the playlist. If the *Repeat* button is enabled, the MP3 playlist is played in a loop.

Table 3.8: Audio input devices for the AAC stream.

Stream Datarate

The *Stream bandwidth* input field enables to adjust the number of bytes reserved for this stream. Spark allows to exceed the number of free MSC bytes in the configuration. However, it is recommended to adjust the number of bytes to be less or equal to the number of free MSC bytes.

The *High protected* input field lets the user choose the percentage of high protected bytes with respect to the bandwidth of the stream. For this high protected part (HPP), a separate protection level can be chosen in the the *DRM Settings* options of the *Transmitter* tab. It is recommended to choose values between 0 and 50 percent for the higher protected part.

Text Messages

DRM allows for text message transmission in parallel to an audio broadcast. In the text message submenu a list of text messages can be defined which are transmitted in the order in which they occur in the list.

A text message is divided into headline and body. The maximum length of a text message (headline and body) is 128 bytes, whereas the number of usable characters varies with respect to the characters used due to the unicode transformation format (UTF) coding.

To create a new text message, the user needs to click on the *NEW* button in the text message submenu and select the *Text Message* option until the text message input dialog appears as shown in Figure 3.13. The new text message is inserted after the selected row in the list. Besides the header and body input fields the user can choose a repetition count from the *Repeat* combo box. If the repetition count is four, the text message is transmitted four times before the next message from the text message queue will be transmitted. This allows the user to coarsely define the display duration of the text message at the receiver. However, the display duration of the current text message depends on the length of the **subsequent** text message in the queue, i.e. if the subsequent text message in the queue is short in terms of number of bytes, the current text message is shown for a shorter time.

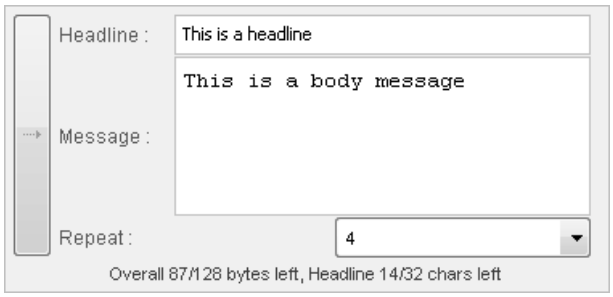

Figure 3.13: The text message input dialog in the AAC configuration panel.

Additional to text messages it is possible add clear display commands by selecting *Clear display command* after pressing the *NEW* button in the text message submenu. This command advises the receiver to clear the last message from the display before the new message will be shown.

3.3.3 PRBS Stream

The PRBS stream configuration comprises the stream bandwidth settings and the PRBS settings as depicted in Figure 3.14. In the PRBS settings it can be specified whether the transmitted sequence shall be synchronized or not. Synchronized means that the sequence generator is reset into its initial state ($0xFFFFFFFF_{HEX}$) at the beginning of every transmission super frame. If the PRBS is not synchronized, the binary sequence will repeat after approximately 2^{23} bits (see [2]).

In the *Stream bandwidth* submenu the number of high protected bytes and the number of low protected bytes for this stream can be defined separately.

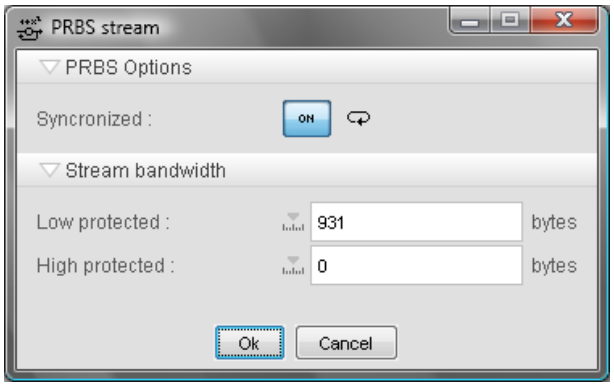

Figure 3.14: The configuration dialog of a PRBS data stream.

3.3.4 Packet Datastream

A packet datastream carries the data for a MOT Slideshow or a MOT Website substream. DRM allows up to four different substreams per datastream whereas each substream is uniquely identified by a *packet ID* [6].

The configuration of a packet datastream is shown in Figure 3.15. The number of low protected bytes of the packet datastream can be adjusted in the *Low protected* input field and the number of high protected bytes can be set in the *High protected* input field.

In the *Packetsize* input field the number of bytes per data packet can be defined. One data packet must not exceed the size of 256 bytes and should not set to be less than 64 bytes for efficiency reasons. After the number of bytes per packet has been changed, the number of available packets in the substream

changes as well. It is recommended to adjust the number of bytes per packet such that it becomes an integer fraction of the number of bytes of the packet stream which is the sum of the number of lower and higher protected bytes.

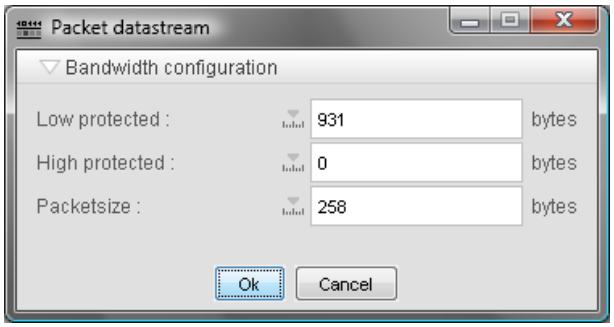

Figure 3.15: The packet datastream configuration dialog.

3.3.5 Packet Datastream / MOT Slideshow Substream

The MOT [7] slideshow substream allows to transmit pictures in a sequential fashion using an image carousel.

A new substream can be created by either clicking on the *Slideshow* button on the left hand side of the content manager panel or by selecting the *Create new substream* option from the packet datastream popup menu and choosing *MOT slideshow* from the application type input box.

Figure 3.16 shows the MOT slideshow configuration dialog, which is subpartitioned into the *Packet configuration* and *Carousel configuration* input panels.

Packet configuration

In the packet configuration panel the number of packets used for this stream can be specified. The maximum number of packets depends on the following items:

- The packet datastream bandwidth (high and low protected parts).
- The number of bytes per packet.
- The number of packets assigned to other substreams of the same packet datastream.

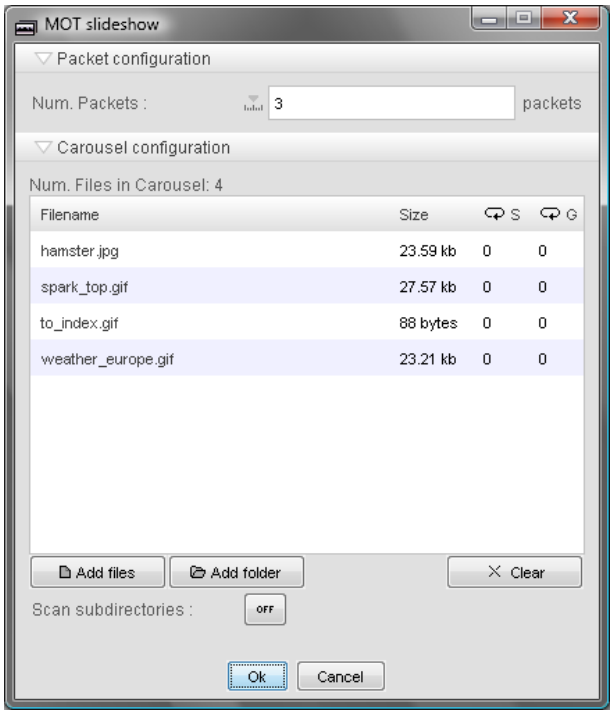

Figure 3.16: MOT slideshow substream configuration dialog.

Hence, the maximum number of packets can be increased by decreasing the number of bytes per packet (packet datastream config dialog), increasing the packet datastream bandwidth (packet datastream config dialog), or decreasing the number of packets that have been assigned to other substreams (substream config dialog) of sharing the same packet datastream.

Carousel configuration

Single images can be added to the MOT carousel by pressing the *Add file* button. The *Add folder* button enables to add multiple of images from a folder to the carousel. When enabling the *Scan subdirectories* button the software scans the subdirectories of the folder that has been selected in the *Add folder* file chooser dialog. After the transmission has been started the images are sequentially transmitted as they appear in the carousel list. Pressing the *Clear* button deletes all entries in the MOT carousel.

The MOT carousel contains filename and filesize of the images. The combo boxes S and G define the repetition rate on the segment layer (S) and on the group-layer (G), respectively. The repetition behavior on the different layers will be explained by the following example:

Given the assumption that the input file is split into three segments A, B and C, whereas all segments of the file are referred to as a group. Then, the segment transmission pattern with a single segment layer repetition would be A, A, B, B, C, C . The transmission pattern with a single group layer repetition would be A, B, C, A, B, C. Hence, the transmission patter with a single segment layer and a single group layer repetition would be $A, A, B, B, C, C, A, A, B, B, C, C$.

3.3.6 Packet Datastream / MOT Website Substream

The MOT website substream allows for the transmission of static web content including pictures, stylesheets and other files. A new MOT website substream can be created by either clicking on the *Website* button on the left hand side of the content manager panel or by selecting the *Create new substream* option from the packet datastream popup menu and choosing *MOT website* from the application type input box. The MOT website configuration dialog is depicted in Figure 3.17.

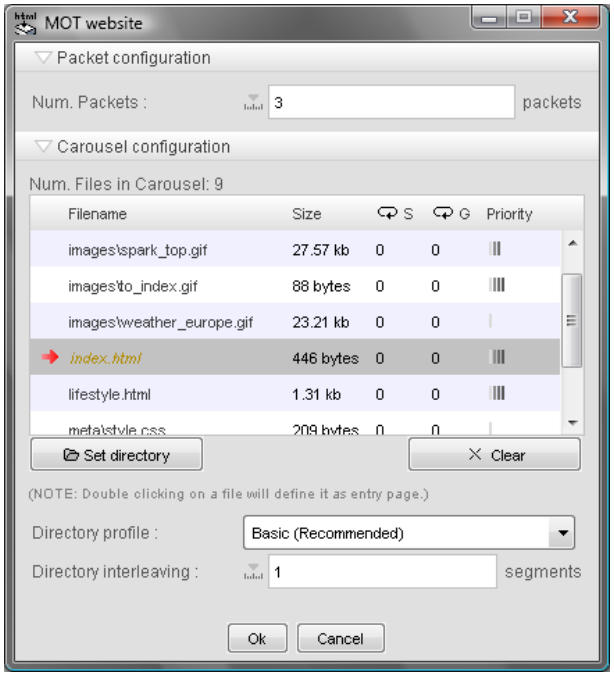

Figure 3.17: MOT website substream configuration dialog.

All files of the website must share the same root directory. The root directory can be defined by selecting a physical directory from the file chooser dialog after pressing the *Set directory* button. After the directory has been loaded to the carousel the filenames are shown in the configuration panel. The *Packet configuration* and the file repetition options for the MOT website substream are equal

to the MOT slideshow configuration and are explained in Section 3.3.5. The transmission priority of a file can be defined by adjusting the *Priority* parameter. A higher transmission priority means a more frequent repetition of the file.

NOTE: For a fast carousel acquisition at the receiver it is recommended to keep the number of files and their respective file sizes as small as possible.

Directory profile

The user can choose between the *Basic* profile and the *Unrestricted PC* profile. In the basic profile the hardware limitations of the receiver, such as screen resolution and color depth, should be considered. It is recommended to design the website for a screen resolution of not more than 320x200 pixels. For more information on the different directory profiles refer to [8].

Directory interleaving

In addition to the website files, directory information (e.g. the filenames, the root directory, etc.) is transmitted in the file carousel during a broadcast. This directory information is transmitted in segments in parallel to the website file segments. The *Directory interleaving* parameter defines the number of directory information segments per website file segment. Smaller numbers guarantee a higher directory information repetition and hence a faster and more robust data acquisition at the receiver.

Setting the directory index

The directory index file is defined to be shown at the receiver after the MOT directory content has been received. The directory index file is similar to the file (commonly in HTML) that is shown in your browser after entering a certain web address. A common name for such an index file is "index.htm". A file from the MOT carousel can be defined as directory index file by **double clicking** on it in the carousel list of the configuration dialog.

3.3.7 Services

Only the service information carried by the services is visible to the user of a DRM receiver. The service information is transmitted over the SDC (see Figure 1.1). Each service is associated with one particular stream in the DRM multiplex. By selecting a service at the receiver, the receiver is advised to decode the stream information that is associated with this service. Hence, a DRM service

can be interpreted as a pointer to a stream with additional side information.

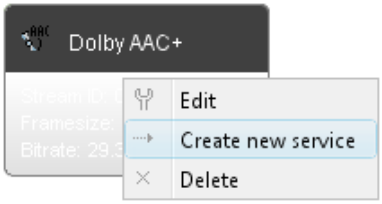

Figure 3.18: Create a new service pointing to an existing Dolby AAC+ audio stream.

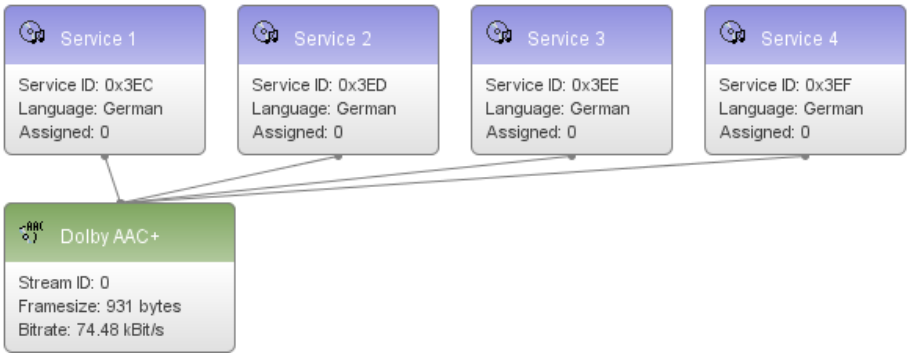

Figure 3.19: Four services pointing to the same stream.

In DRM, multiple services can point to the same stream (rule 1) as shown in Figure 3.19, whereas it is permitted to have multiple streams pointing to the same service (rule 2). However, there is one exception to the second rule: If there exist two streams of which one is an audio stream and the other one is a data stream, a service is allowed to be assigned to both of the streams as depicted in Figure 3.20. In this case, although the service carries *audio and data*, the receiver refers to the service as an audio service with additional data.

In DRM the maximum number of services that can be created and assigned is limited to four services. Each service is uniquely identified by a *short ID* from 0 to 3.

In order to add a new service the content manager must contain at least one stream to which the new service can be assigned and there must be less than four existing services, i.e. the maximum number of services must not be exceeded.

A new service can be created by right clicking on an existing stream (green box) and choosing *Create new service* from the dialog as shown in Figure 3.18.

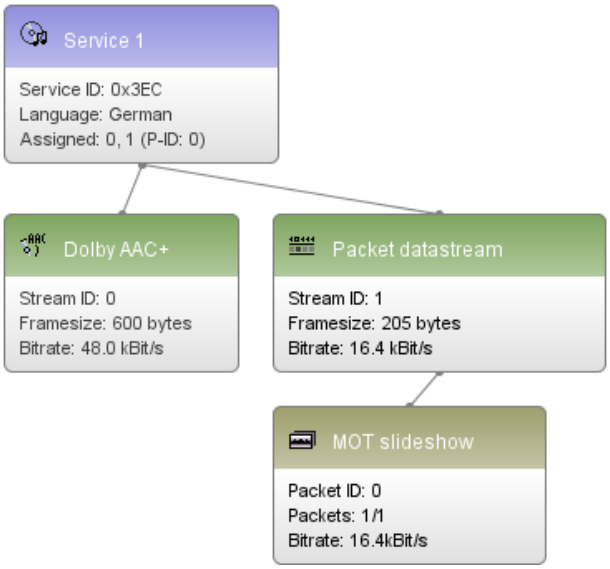

Figure 3.20: A service pointing to an audio and a data stream at the same time.

The new service is automatically assigned to the stream from which it has been created. To change the assignment the user must click on the service and select *Assign to stream* from the popup menu. Figure 3.21 shows a reassignment of a service which originally pointed to a stream with stream ID 0 and which is being readjusted to point to a stream with stream ID 1.

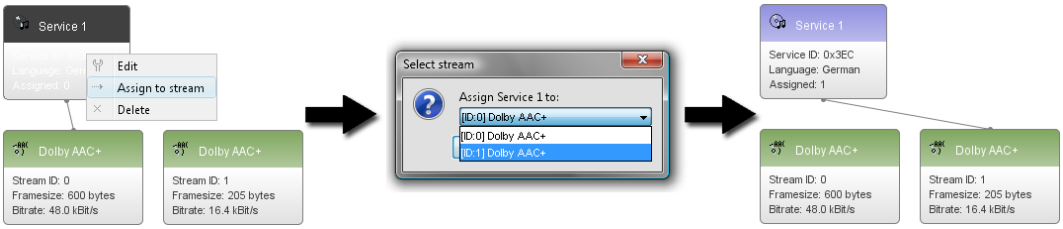

Figure 3.21: Change the service-to-stream assignment.

After the service has been created the user can change the service parameters by right clicking on the new service (blue box) and selecting *Edit* from the input dialog as depicted in Figure 3.22.

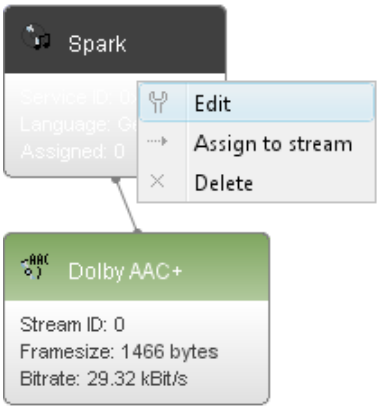

Figure 3.22: Edit the service parameters.

After the *Edit* option has been selected the service parameter window appears as shown in Figure 3.23. In the following sections the different service parameters are briefly explained.

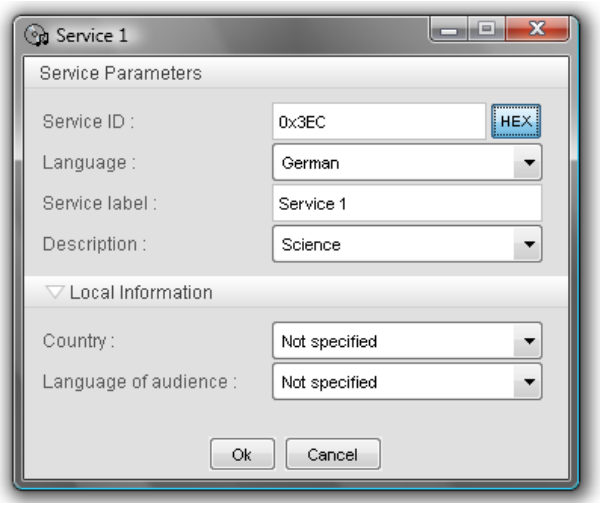

Figure 3.23: The service parameters.

Service ID

The service identifier is a 24 bit number which uniquely identifies a service. Typically, the number is shown and entered in hexadecimal notation.

Language

The *Language* input field defines the language of the stream where this service is pointing to. The language information is transmitted in the SDC service param-

eters over the FAC.

Service label

The service label describes the name of the service with up to 16 characters (e.g. "Classic-Radio"). The service label is transmitted over the SDC in the *Label data entity*.

Description

There exist 32 different service descriptions which describe the content of the stream the service is pointing to. The service description is transmitted in the SDC service parameters over the FAC.

Local Information / Country

The user can choose the country from which this broadcast is transmitted. The country information is transmitted over the SDC in the *Language and country data entity*.

Local Information / Language of audience

The user can choose the audience language more specifically. This language information is transmitted over the SDC in the *Language and country data entity*.

3.4 The Time Reference Settings

In the *Time* tab two network time protocol (NTP) servers as well as the local Greenwich mean time (GMT) offset can be specified as a time reference for the broadcast. In addition to the NTP server and the GMT input fields the NTP poll interval in milliseconds can be specified. A screenshot of the time tab is shown in Figure 3.24.

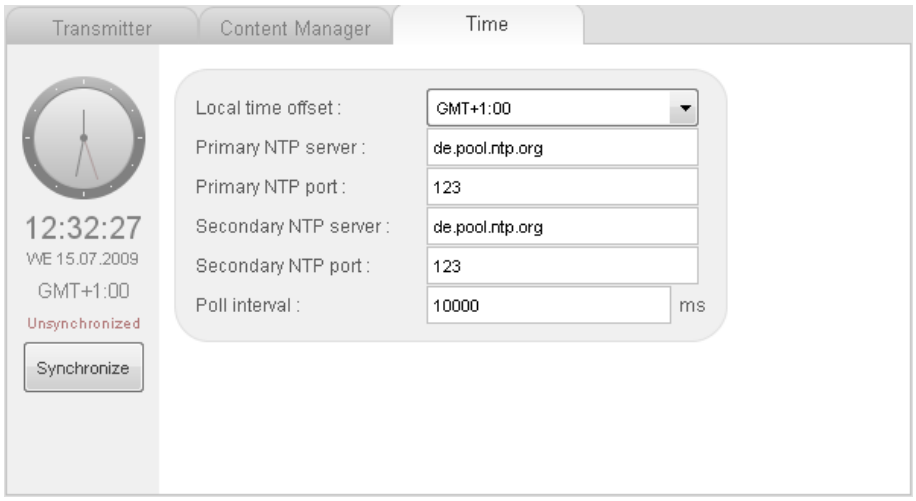

Figure 3.24: The time reference tab.

On the left hand side, the panel shows the internal **system time and date** and the NTP synchronization status. Spark can be configured to automatically synchronize the system time with the NTP server specified in the input fields on startup. A successful synchronization is indicated by a green "synchronized" label in the synchronization status panel. In case the NTP synchronization failed, the label shows the word "unsynchronized" in red letters. By pressing the *Synchronize* button in the same panel, the user can retry to synchronize the time by polling the respective NTP servers.

During the transmission the application continuously synchronizes the system clock in periods of the poll interval. The poll interval can be defined in the *Poll interval* input field.

The synchronization procedure works as follows:

1. Poll the 1st NTP server.
2. If a timestamp from the 1st NTP server has been received → Synchronize with the received timestamp and exit.
3. Otherwise, poll the 2nd NTP server.
4. If a timestamp from the 2nd NTP server has been received → Synchronize with the received timestamp and exit.
5. Otherwise → Exit.

NOTE: "Unsynchronized" means that Spark uses the CMOS time of the PC BIOS as time reference.

3.5 Control Panel

The control panel is positioned on the right hand side of the Spark application window as depicted in Figure 3.1. The panel contains several buttons to manage the operation of the application. In the next sections the functionality of the different buttons will be explained.

3.5.1 Starting and Stopping the Transmission

After the application configuration has been finished, the transmission can be started by pressing the start button as shown in 3.26.

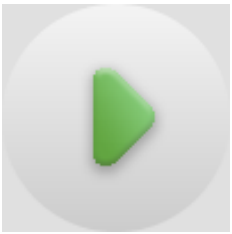

Figure 3.25: Button to start the transmission.

If there are no configuration errors, the output status window appears and the start button changes its appearance and becomes a stop button with flashing green backlight as depicted in Figure **??**.

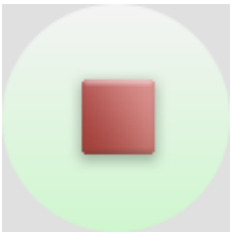

Figure 3.26: Button to stop the transmission.

3.5.2 Open and Save the Application Settings

The changes that have been made in the configuration window can be loaded and saved by pressing the respective open and save buttons (Figure 3.27). A successful parameter reading and writing is indicated by a short green flashing of the background of the open or save buttons. Figure 3.28 shows the mentioned behavior for the open button.

Figure 3.27: Buttons to open and save the application settings.

Figure 3.28: A successful read operation is indicated by a green flashing background of the open button.

3.5.3 Text Message Reconfiguration

Text messages can be entered in the AAC audio stream configuration dialog as explained in Section 3.3.2. After the transmission has been started it is possible to reconfigure the text messages during the broadcast by pressing the text message reconfiguration button which is shown in Figure 3.29. After the text message reconfiguration has been finished successfully, the reconfiguration button flashes green.

NOTE: A text message reconfiguration requires the stream configuration in the content manager to remain unchanged during the transmission. Furthermore, text message reconfiguration will not function if the text message queue of an audio stream had been empty at the time the transmission had been started.

Figure 3.29: The text message reconfiguration button.

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